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FORM PTO-1390 (Modified) (REV 10-95) U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE		ATTORNEY'S DOCKET NUMBER
TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371		62-194
		U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR 09/367153
INTERNATIONAL APPLICATION NO. **PCT/GB98/03714**	INTERNATIONAL FILING DATE **11 December 1998 (11.12.98)**	PRIORITY DATE CLAIMED **13 December 1997 (13.12.97)**

TITLE OF INVENTION
A METHOD OF PROCESSING AN AUDIO SIGNAL

APPLICANT(S) FOR DO/EO/US
Alastair Sibbald, Fawad Nackvi and Richard David Clemow

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).
4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.
5. ☒ A copy of the International Application as filed (35 U.S.C. 371 (c) (2))
   a. ☐ is transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☒ has been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☐ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
7. ☐ A copy of the International Search Report (PCT/ISA/210).
8. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371 (c)(3))
   a. ☐ are transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ have been transmitted by the International Bureau.
   c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   d. ☒ have not been made and will not be made.
9. ☐ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
10. ☒ An oath or declaration of the inventor(s) (35 U.S.C. 371 (c)(4)).
11. ☐ A copy of the International Preliminary Examination Report (PCT/IPEA/409).
12. ☐ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371 (c)(5)).

Items 13 to 18 below concern document(s) or information included:

13. ☐ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
14. ☒ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
15. ☒ A **FIRST** preliminary amendment.
    A **SECOND** or **SUBSEQUENT** preliminary amendment.
16. ☐ A substitute specification.
17. ☐ A change of power of attorney and/or address letter.
18. ☐ Certificate of Mailing by Express Mail
19. ☐ Other items or information:

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U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR) 09/367153	INTERNATIONAL APPLICATION NO. PCT/GB98/03714	ATTORNEY'S DOCKET NUMBER 62-194

20. The following fees are submitted:.

BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)) :

		CALCULATIONS	PTO USE ONLY
☒ Search Report has been prepared by the EPO or JPO	**$840.00**		
☐ International preliminary examination fee paid to USPTO (37 CFR 1.482)	**$670.00**		
☐ No international preliminary examination fee paid to USPTO (37 CFR 1.482) but international search fee paid to USPTO (37 CFR 1.445(a)(2))	**$760.00**		
☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2) paid to USPTO	**$970.00**		
☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(2)-(4)	**$96.00**		
ENTER APPROPRIATE BASIC FEE AMOUNT =		**$840.00**	
Surcharge of **$130.00** for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (e)).		**$0.00**	

CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE		
Total claims	40 - 20 =	20	x $18.00	**$360.00**	
Independent claims	4 - 3 =	1	x $78.00	**$78.00**	
Multiple Dependent Claims **(check if applicable)**.			☐	**$0.00**	
TOTAL OF ABOVE CALCULATIONS =				**$1,278.00**	
Reduction of 1/2 for filing by small entity, if applicable. Verified Small Entity Statement must also be filed (Note 37 CFR 1.9, 1.27, 1.28) **(check if applicable)**.			☐	**$0.00**	
SUBTOTAL =				**$1,278.00**	
Processing fee of **$130.00** for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (f)).			+	**$0.00**	
TOTAL NATIONAL FEE =				**$1,278.00**	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31) **(check if applicable)**.			☒	**$40.00**	
TOTAL FEES ENCLOSED =				**$1,318.00**	
				Amount to be: refunded	$
				charged	$

☒ A check in the amount of **$1,318.00** to cover the above fees is enclosed.

☐ Please charge my Deposit Account No. in the amount of to cover the above fees. A duplicate copy of this sheet is enclosed.

☒ The Commissioner is hereby authorized to charge any fees which may be required, or credit any overpayment to Deposit Account No. **50-0687** A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

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Will H. Boll
SIGNATURE

William H. Bollman
NAME

36,457
REGISTRATION NUMBER

Augsut 9, 1999
DATE

09/367153
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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of:

SIBBALD et al.

Serial No.: **Unassigned** Examiner: **Unassigned**

Filed: Concurrently Herewith Group Art Unit: **Unassigned**

Title: **A METHOD OF PROCESSING AN AUDIO SIGNAL**

Honorable Commissioner of Patents and Trademarks
Washington, D.C. 20231

August 9, 1999

PRELIMINARY AMENDMENT

Dear Sir:

Prior to action in, and calculation of fees for, the above captioned application, please amend the application as follows:

IN THE CLAIMS:

Kindly cancel claims 1-13, and add the following new claims 14-53:

---14. (New) A method of processing a single channel audio signal corresponding to a sound from a sound source located at a source position relative to a preferred position of a listener, comprising:

providing a right channel and a left channel, each of said right channel and said left channel carrying said single channel audio signal;

modifying said single channel audio signal of each of said right channel and said left channel using at least one of a plurality of head response transfer functions to provide a right signal in said right channel for a right ear of said listener and a left signal in said left channel for a left ear of said listener;

introducing a time delay between said right channel and said left channel to provide cues to perception of a direction of said source position relative to said preferred position of said listener at a given time, said time delay corresponding to an inter-aural time difference of said sound from said sound source with respect to said listener; and

choosing respective values for magnitude of said left signal and magnitude of said right signal to provide cues for perception of a distance of said source position from said preferred position at said given time.

15. (New) The method of processing a single channel audio signal in accordance with claim 14, wherein:

said respective values for magnitude of said left signal and said magnitude of said right signal are chosen separately.

16. (New) The method of processing a single channel audio signal in accordance with claim 14, wherein:

said respective values for magnitude of said left signal and said magnitude of said right signal are determined based on an inverse of square of a distance between said source position and respective ears of said listener.

17. (New) The method of processing a single channel audio signal in accordance with claim 14, wherein:

said step of choosing respective values for magnitude of said left signal and magnitude of said right signal comprises:

providing a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal; and

selecting said values for magnitude from said look-up table.

18. (New) The method of processing a single channel audio signal in accordance with claim 16, wherein:

said step of choosing respective values for magnitude of said left signal and magnitude of said right signal comprises:

selecting a distance from said source position to a center of a head of said listener at said given time; and

determining said distance between said source position and respective ears of said listener based on said inter-aural time delay.

19. (New) The method of processing a single channel audio signal in accordance with claim 18, wherein:

said step of choosing respective values for magnitude of said left signal and magnitude of said right signal further comprises:

providing a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal; and

selecting said values for magnitude from said look-up table.

20. (New) The method of processing a single channel audio signal in accordance with claim 14, wherein:

at least one of said magnitude of said left signal and said magnitude of said right signal is sufficiently small as to be inaudible.

21. (New) The method of processing a single channel audio signal in accordance with claim 14, wherein:

said left signal and said right signal are compensated to provide at least one of a cancellation and a reduction of transaural crosstalk when said left signal and said right signal are supplied through said left channel and said right channel respectively for replay by loudspeakers.

22. (New) The method of processing a single channel audio signal in accordance with claim 14, further comprising:

combining said left signal and said right signal with other two or more channel audio signals.

23. (New) The method of processing a single channel audio signal in accordance with claim 22, wherein:

said step of combining comprises:

adding respective contents of said left channel and said right channel to corresponding channels of said other two or more channel signals.

24. (New) A computer readable storage medium having stored thereon a computer program for implementing a method of processing a single channel audio signal corresponding to a sound from a sound source located at a source position relative to a preferred position of a listener, said computer program comprising a set of instructions for:

providing a right channel and a left channel, each of said right channel and said left channel carrying said single channel audio signal;

modifying said single channel audio signal of each of said right channel and said left channel using at least one of a plurality of head response transfer functions to provide a right signal in said right channel for a right ear of said listener and a left signal in said left channel for a left ear of said listener;

introducing a time delay between said right channel and said left channel to provide cues to perception of a direction of said source position relative to said preferred position of said listener at a given time, said time delay corresponding to an inter-aural time difference of said sound from said sound source with respect to said listener; and

choosing respective values for magnitude of said left signal and magnitude of said right signal to provide cues for perception of a distance of said source position from said preferred position at said given time.

25. (New) The computer readable storage medium according to claim 24, wherein:

said respective values for magnitude of said left signal and said magnitude of said right signal are chosen separately.

26. (New) The computer readable storage medium according to claim 24, wherein:

said respective values for magnitude of said left signal and said magnitude of said right signal are determined based on an inverse of square of a distance between said source position and respective ears of said listener.

27. (New) The computer readable storage medium according to claim 24, wherein:

said set of instructions for choosing respective values for magnitude of said left signal and magnitude of said right signal comprises a set of instructions for:

providing a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal; and

selecting said values for magnitude from said look-up table.

28. (New) The computer readable storage medium according to claim 26, wherein:

said set of instructions for choosing respective values for magnitude of said left signal and magnitude of said right signal comprises a set of instructions for:

selecting a distance from said source position to a center of a head of said listener at said given time; and

determining said distance between said source position and respective ears of said listener based on said inter-aural time delay.

29. (New) The computer readable storage medium according to claim 28, wherein:

said set of instructions for choosing respective values for magnitude of said left signal and magnitude of said right signal further comprises a set of instructions for:

providing a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal; and

selecting said values for magnitude from said look-up table.

30. (New) The computer readable storage medium according to claim 24, wherein:

at least one of said magnitude of said left signal and said magnitude of said right signal is sufficiently small as to be inaudible.

31. (New) The computer readable storage medium according to claim 24, wherein:

said left signal and said right signal are compensated to provide at least one of a cancellation and a reduction of transaural crosstalk when said left signal and said right signal are supplied through said left channel and said right channel respectively for replay by loudspeakers.

32. (New) The computer readable storage medium according to claim 24, wherein:

said computer program further comprises a set of instructions for:

combining said left signal and said right signal with other two or more channel audio signals.

33. (New) The computer readable storage medium according to claim 32, wherein:

said set of instructions for combining said left signal and said right signal with other two or more channel audio signals comprises a set of instructions for:

adding respective contents of said left channel and said right channel to corresponding channels of said other two or more channel signals.

34. (New) An apparatus for processing a single channel audio signal corresponding to a sound from a sound source located at a source position relative to a preferred position of a listener, comprising:

means for providing a right channel and a left channel, each of said right channel and said left channel adapted to carry said single channel audio signal;

means for modifying said single channel audio signal of each of said right channel and said left channel using at least one of a plurality of head response transfer functions to provide a right signal in said right channel for a right ear of said listener and a left signal in said left channel for a left ear of said listener;

means for introducing a time delay between said right channel and said left channel to provide cues to perception of a direction of said source position relative to said preferred position of said listener at a given time, said time delay corresponding to an inter-aural time difference of said sound from said sound source with respect to said listener; and

means for choosing respective values for magnitude of said left signal and magnitude of said right signal to provide cues for perception of a distance of said source position from said preferred position at said given time.

35. (New) The apparatus for processing a single channel audio signal according to claim 34, wherein:

said means for choosing said respective values is adapted to choose said respective values for magnitude of said left signal and said magnitude of said right signal separately.

36. (New) The apparatus for processing a single channel audio signal according to claim 34, wherein:

said means for choosing said respective values is adapted to choose said respective values for magnitude of said left signal and said magnitude of said right signal based on an inverse of square of a distance between said source position and respective ears of said listener.

37. (New) The apparatus for processing a single channel audio signal according to claim 34, wherein:

said means for choosing respective values for magnitude of said left signal and magnitude of said right signal comprises:

a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal; and

means for selecting said values for magnitude from said look-up table.

38. (New) The apparatus for processing a single channel audio signal according to claim 36, wherein:

said means for choosing respective values for magnitude of said left signal and magnitude of said right signal comprises:

means for selecting a distance from said source position to a center of a head of said listener at said given time; and

means for determining said distance between said source position and respective ears of said listener based on said inter-aural time delay.

39. (New) The apparatus for processing a single channel audio signal according to claim 37, wherein:

said means for choosing respective values for magnitude of said left signal and magnitude of said right signal further comprises:

a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal; and

means for selecting said values for magnitude from said look-up table.

40. (New) The apparatus for processing a single channel audio signal according to claim 34, wherein:

at least one of said magnitude of said left signal and said magnitude of said right signal is sufficiently small as to be inaudible.

41. (New) The apparatus for processing a single channel audio signal according to claim 34, further comprising:

compensating means for providing at least one of a cancellation and a reduction of transaural crosstalk in said left signal and said right signal when said left signal and said right signal are supplied through said left channel and said right channel respectively for replay by loudspeakers.

42. (New) The apparatus for processing a single channel audio signal according to claim 34, further comprising:

means for combining said left signal and said right signal with other two or more channel audio signals.

43. (New) The apparatus for processing a single channel audio signal according to claim 42, wherein:

said means for combining comprises:

means for adding respective contents of said left channel and said right channel to corresponding channels of said other two or more channel signals.

44. (New) An audio signal, comprising:

a right signal for a right ear of a listener, said right signal being obtained by modifying a single channel audio signal using at least one of a plurality of head response transfer functions, said single channel audio signal corresponding to a sound from a sound source located at a source position relative to a preferred position of said listener; and

a left signal for a left ear of said listener, said left signal being obtained by modifying said single channel audio signal using at least one of a plurality of head response transfer functions,

wherein said left signal and said right signal having therebetween a time delay to provide cues to perception of a direction of said source position relative to said preferred position of said listener at a given time, said time delay corresponding to an inter-aural time difference of said sound from said sound source with respect to said listener; and

wherein respective values for magnitude of said left signal and magnitude of said right signal is chosen to provide cues for perception of a distance of said source position from said preferred position at said given time.

45. (New) The audio signal according to claim 44, wherein:

said respective values for magnitude of said left signal and said magnitude of said right signal are chosen separately.

46. (New) The audio signal according to claim 44, wherein:

said respective values for magnitude of said left signal and said magnitude of said right signal are determined based on an inverse of square of a distance between said source position and respective ears of said listener.

47. (New) The audio signal according to claim 44, wherein:

said respective values for magnitude of said left signal and magnitude of said right signal are chosen by selecting said values for magnitude from a look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal.

48. (New) The audio signal according to claim 46, wherein:

said respective values for magnitude of said left signal and magnitude of said right signal are chosen by selecting a distance from said source position to a center of a head of said listener at said given time, and by determining said distance between said source position and respective ears of said listener based on said inter-aural time delay.

49. (New) The audio signal according to claim 48, wherein:

said respective values for magnitude of said left signal and magnitude of said right signal are chosen by selecting said values for magnitude from said look-up table having thereon distances between said source position and respective ears of said listener, said distances corresponding to associative ones of said values for magnitude of said left signal and said magnitude of said right signal.

50. (New) The audio signal according to claim 44, wherein:

at least one of said magnitude of said left signal and said magnitude of said right signal is sufficiently small as to be inaudible.

51. (New) The audio signal according to claim 44, wherein:

said left signal and said right signal are compensated to provide at least one of a cancellation and a reduction of transaural crosstalk when said left signal and said right signal are supplied through said left channel and said right channel respectively for replay by loudspeakers.

52. (New) The audio signal according to claim 44, wherein:

said left signal and said right signal are combined with other two or more channel audio signals.

53. (New) The audio signal according to claim 52, wherein:

said left signal and said right signal are combined by adding respective contents of said left channel and said right channel to corresponding channels of said other two or more channel signals.----

REMARKS

By this amendment, claims 1-13 are canceled, and replaced with new claims 14-53. Thus, claims 14-53 are currently pending in the present application.

Early and favorable consideration of the present application is earnestly solicited.

Respectfully submitted,

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A METHOD OF PROCESSING AN AUDIO SIGNAL

This invention relates to a method of processing a single channel audio signal to provide an audio signal having left and right channels corresponding to a sound source at a given direction in space relative to a preferred position of a listener in use, the information in the channels including cues for perception of the direction of said single channel audio signal from said preferred position, the method including the steps of: a) providing a two channel signal having the same single channel signal in the two channels; b) modifying the two channel signal by modifying each of the channels using one of a plurality of head response transfer functions to provide a right signal in one channel for the right ear of a listener and a left signal in the other channel for the left ear of the listener; and c) introducing a time delay between the channels corresponding to the inter-aural time difference for a signal coming from said given direction, the inter-aural time difference providing cues to perception of the direction of the sound source at a given time.

The processing of audio signals to reproduce a three dimensional sound-field on replay to a listener having two ears has been a goal for inventors since the invention of stereo by Alan Blumlein in the 1930's. One approach has been to use many sound reproduction channels to surround the listener with a multiplicity of sound sources such as loudspeakers. Another approach has been to use a dummy head having microphones positioned in the auditory canals of artificial ears to make sound recordings for headphone listening. An especially promising approach to the binaural synthesis of such a sound-field has been described in EP-B-0689756, which describes the synthesis of a sound-field using a pair of loudspeakers and only two signal channels, the sound-field nevertheless having directional information allowing a listener to perceive sound sources appearing to lie anywhere on a sphere surrounding the head of a listener placed at the centre of the sphere.

A drawback with such systems developed in the past has been that although the recreated sound-field has directional information, it has been difficult to recreate the perception of having a sound source which is close to the

listener, typically a source which appears to be closer than about 1.5 metres from the head of a listener. Such sound effects would be very effective for computer games for example, or any other application when it is desired to have sounds appearing to emanate from a position in space close to the head of a listener, or a sound source which is perceived to move towards or away from a listener with time, or to have the sensation of a person whispering in the listener's ear.

According to a first aspect of the invention there is provided a method as specified in claims 1 - 11. According to a second aspect of the invention there is provided apparatus as specified in claim 12. According to a third aspect of the invention there is provided an audio signal as specified in claim 13.

Embodiments of the invention will now be described, by way of example only, with reference to the accompanying diagrammatic drawings, in which
Figure 1 shows the head of a listener and a co-ordinate system,
Figure 2 shows a plan view of the head and an arriving sound wave,
Figure 3 shows the locus of points having an equal inter-aural or inter-aural time delay,
Figure 4 shows an isometric view of the locus of Figure 3,
Figure 5 shows a plan view of the space surrounding a listener's head,
Figure 6 shows further plan views of a listener's head showing paths for use in calculations of distance to the near ear,
Figure 7 shows further plan views of a listener's head showing paths for use in calculations of distance to the far ear,
Figure 8 shows a block diagram of a prior art method,
Figure 9 shows a block diagram of a method according to the present invention,
Figure 10 shows a plot of near ear gain as a function of azimuth and distance, and
Figure 11 shows a plot of far ear gain as a function of azimuth and distance.

The present invention relates particularly to the reproduction of 3D-sound from two-speaker stereo systems or headphones. This type of 3D-sound is described, for example, in EP-B-0689756 which is incorporated herein by reference.

It is well known that a mono sound source can be digitally processed via a pair of "Head-Response Transfer Functions" (HRTFs), such that the resultant stereo-pair signal contains 3D-sound cues. These sound cues are introduced naturally by the head and ears when we listen to sounds in real life, and they include the inter-aural amplitude difference (IAD), inter-aural time difference (ITD) and spectral shaping by the outer ear. When this stereo signal pair is introduced efficiently into the appropriate ears of the listener, by headphones say, then he or she perceives the original sound to be at a position in space in accordance with the spatial location of the HRTF pair which was used for the signal-processing.

When one listens through loudspeakers instead of headphones, then the signals are not conveyed efficiently into the ears, for there is "transaural acoustic crosstalk" present which inhibits the 3D-sound cues. This means that the left ear hears a little of what the right ear is hearing (after a small, additional time-delay of around 0.2 ms), and vice versa. In order to prevent this happening, it is known to create appropriate "crosstalk cancellation" signals from the opposite loudspeaker. These signals are equal in magnitude and inverted (opposite in phase) with respect to the crosstalk signals, and designed to cancel them out. There are more advanced schemes which anticipate the secondary (and higher order) effects of the cancellation signals themselves contributing to secondary crosstalk, and the correction thereof, and these methods are known in the prior art.

When the HRTF processing and crosstalk cancellation are carried out correctly, and using high quality HRTF source data, then the effects can be quite remarkable. For example, it is possible to move the virtual image of a sound-source around the listener in a complete horizontal circle, beginning in front, moving around the right-hand side of the listener, behind the listener, and back around the left-hand side to the front again. It is also possible to make the sound source move in a vertical circle around the listener, and indeed make the sound appear to come from any selected position in space. However, some particular positions are more difficult to synthesise than others, some for psychoacoustic reasons, we believe, and some for practical reasons.

For example, the effectiveness of sound sources moving directly upwards and downwards is greater at the sides of the listener (azimuth = 90°) than directly in front (azimuth = 0°). This is probably because there is more left-right difference information for the brain to work with. Similarly, it is difficult to differentiate between a sound source directly in front of the listener (azimuth = 0°) and a source directly behind the listener (azimuth = 180°). This is because there is no time-domain information present for the brain to operate with (ITD = 0), and the only other information available to the brain, spectral data, is similar in both of these positions. In practice, there is more HF energy perceived when the source is in front of the listener, because the high frequencies from frontal sources are reflected into the auditory canal from the rear wall of the concha, whereas from a rearward source, they cannot diffract around the pinna sufficiently to enter the auditory canal effectively.

In practice, it is known to make measurements from an artificial head in order to derive a library of HRTF data, such that 3D-sound effects can be synthesised. It is common practice to make these measurements at distances of 1 metre or thereabouts, for several reasons. Firstly, the sound source used for such measurements is, ideally, a point source, and usually a loudspeaker is used. However, there is a physical limit on the minimum size of loudspeaker diaphragms. Typically, a diameter of several inches is as small as is practical whilst retaining the power capability and low-distortion properties which are needed. Hence, in order to have the effects of these loudspeaker signals representative of a point source, the loudspeaker must be spaced at a distance of around 1 metre from the artificial head. Secondly, it is usually required to create sound effects for PC games and the like which possess apparent distances of several metres or greater, and so, because there is little difference between HRTFs measured at 1 metre and those measured at much greater distances, the 1 metre measurement is used.

The effect of a sound source appearing to be in the mid-distance (1 to 5 m, say) or far-distance (>5 m) can be created easily by the addition of a reverberation signal to the primary signal, thus simulating the effects of reflected sound waves

from the floor and walls of the environment. A reduction of the high frequency (HF) components of the sound source can also help create the effect of a distant source, simulating the selective absorption of HF by air, although this is a more subtle effect. In summary, the effects of controlling the apparent distance of a sound source beyond several metres are known.

However, in many PC games situations, it is desirable to have a sound effect appear to be very close to the listener. For example, in an adventure game, it might be required for a "guide" to whisper instructions into one of the listener's ears, or alternatively, in a flight-simulator, it might be required to create the effect that the listener is a pilot, hearing air-traffic information via headphones. In a combat game, it might be required to make bullets appear to fly close by the listener's head. These effects are not possible with HRTFs measured at 1 metre distance.

It is therefore desirable to be able to create "near-field" distance effects, in which the sound source can appear to move from the loudspeaker distance, say, up close to the head of the listener, and even appear to "whisper" into one of the ears of the listener. In principle, it might be possible to make a full set of HRTF measurements at differing distances, say 1 metre, 0.9 metre, 0.8 metre and so on, and switch between these different libraries for near-field effects. However, as already noted above, the measurements are compromised by the loudspeaker diaphragm dimensions which depart from point-source properties at these distances. Also, an immense effort is required to make each set of HRTF measurements (typically, an HRTF library might contain over 1000 HRTF pairs which take several man weeks of effort to measure, and then a similar time is required to process these into useable filter coefficients), and so it would be very costly to do this. Also, it would require considerable additional memory space to store each additional HRTF library in the PC. A further problem would be that such an approach would result in quantised-distance effects: the sound source could not move smoothly to the listener's head, but would appear to "jump" when switching between the different HRTF sets.

Ideally, what is required is a means of creating near-field distance effects using a "standard" 1 metre HRTF set.

The present invention comprises a means of creating near-field distance effects for 3D-sound synthesis using a "standard" 1 metre HRTF set. The method uses an algorithm which controls the relative left-right channel amplitude difference as a function of (a) required proximity, and (b) spatial position. The algorithm is based on the observation that when a sound source moves towards the head from a distance of 1 metre, then the individual left and right-ear properties of the HRTF do not change a great deal in terms of their spectral properties. However, their amplitudes, and the amplitude difference between them, do change substantially, caused by a distance ratio effect. The small changes in spectral properties which do occur are related largely to head-shadowing effects, and these can be incorporated into the near-field effect algorithm in addition if desired.

In the present context, the expression "near-field" is defined to mean that volume of space around the listener's head up to a distance of about 1 - 1.5 metre from the centre of the head. For practical reasons, it is also useful to define a "closeness limit", and a distance of 0.2 m has been chosen for the present purpose of illustrating the invention. These limits have both been chosen purely for descriptive purposes, based respectively upon a typical HRTF measurement distance (1 m) and the closest simulation distance one might wish to create, in a game, say. However, it is also important to note that the ultimate "closeness" is represented by the listener hearing the sound ONLY in a single ear, as would be the case if he or she were wearing a single earphone. This, too, can be simulated, and can be regarded as the ultimately limiting case for close to head or "near-field" effects. This "whispering in one ear effect" can be achieved simply by setting the far ear gain to zero, or to a sufficiently low value to be inaudible. Then, when the processed audio signal is is auditioned on headphones, or via speakers after appropriate transaural crosstalk cancellation processing, the sounds appear to be "in the ear".

First, consider for example the amplitude changes. When the sound source moves towards the head from 1 metre distance, the distance ratio (left-ear to sound source vs. right-ear to sound source) becomes greater. For example, for a sound sourcw at 45° azimuth in the horizontal plane, at a distance of 1 metre from the centre of the head, the near ear is about 0.9 metre distance and the far-ear around 1.1 metre. So the ratio is (1.1 / 0.9) = 1.22. When the sound source moves to a distance of 0.5 metre, then the ratio becomes (0.6 / 0.4) = 1.5, and when the distance is 20 cm, then the ratio is approximately (0.4 / 0.1) = 4. The intensity of a sound source diminishes with distance as the energy of the propagating wave is spread over an increasing area. The wavefront is similar to an expanding bubble, and the energy density is related to the surface area of the propagating wavefront, which is related by a square law to the distance travelled (the radius of the bubble).

This gives the well known inverse square law reducion in intensity with distance travelled for a point source. The intensity ratios of left and right channels are related to the inverse ratio of the squares of the distances. Hence, the intensity ratios for distances of 1 m, 0.5 m and 0.2 m are approximately 1.49, 2.25 and 16 respectively. In dB units, these ratios are 1.73 dB, 3.52 dB and 12.04 dB respectively.

Next, consider the head-shadowing effects. When a sound source is 1 metre from the head, at azimuth 45°, say, then the incoming sound waves only have one-quarter of the head to travel around in order to reach the furthermost ear, lying in the shadow of the head. However, when the sound source is much closer, say 20 cm, than the waves have an entire hemisphere to circumnavigate before they can reach the furthermost ear. Consequently, the HF components reaching the furthermost ear are proportionately reduced.

It is important to note, however, that the situation is more complicated than described in the above example, because the intensity ratio differences are position dependent. For example, if the aforementioned situation were repeated for a frontal sound source (azimuth 0°) approaching the head, then there would be

no difference between the left and right channel intensities, because of symmetry. In this instance, the intensity level would simply increase according to the inverse square law.

How then might it be possible to link any particular, close, position in three dimensional space with an algorithm to control the L and R channel gains correctly and accurately? The key factor is the inter-aural time delay, for this can be used to index the algorithm to spatial position in a very effective and efficient manner.

The invention is best described in several stages, beginning with an account of the inter-aural time-delay and followed by derivations of approximate near-ear and far-ear distances in the listener's near-field. Figure 1 shows a diagram of the near-field space around the listener, together with the reference planes and axes which will be referred to during the following descriptions, in which P-P′ represents the front-back axis in the horizontal plane, intercepting the centre of the listener's head, and with Q-Q′ representing the corresponding lateral axis from left to right.

As has already been noted, there is a time-of-arrival difference between the left and right ears when a sound wave is incident upon the head, unless the sound source is in the median plane, which includes the pole positions (i.e. directly in front, behind above and below). This is known as the inter-aural time delay (ITD), and can be seen depicted in diagram form in Figure 2, which shows a plan view of a conceptual head, with left ear and right ear receiving a sound signal from a distant source at azimuth angle θ (about +45° as shown here). When the wavefront (W - W′) arrives at the right ear, then it can be seen that there is a path length of (a + b) still to travel before it arrives at the left ear (LE). By the symmetry of the configuration, the b section is equal to the distance from the head centre to wavefront W - W′, and hence: $b = r.\sin\theta$. It will be clear that the arc a represents a proportion of the circumference, subtended by θ. By inspection, then, the path length (a+b) is given by:

$$\text{path length} = \left(\frac{\theta}{360}\right)2\pi r + r.\sin\theta \qquad (1)$$

(This path length (in cm units) can be converted into the corresponding time-delay value (in ms) by dividing by 34.3.)

It can be seen that, in the extreme, when θ tends to zero, so does the path length. Also, when θ tends to 90°, and the head diameter is 15 cm, then the path length is about 19.3 cm, and the associated ITD is about 563 μs. In practice, the ITDs are measured to be slightly larger than this, typically up to 702 μs. It is likely that this is caused by the non-spherical nature of the head (including the presence of the pinnae and nose), the complex diffractive situation and surface effects.

At this stage, it is important to appreciate that, although this derivation relates only to the front-right quadrant in the horizontal plane (angles of azimuth between 0° and 90°), it is valid in all four quadrants. This is because (a) the front-right and right-rear quadrants are symmetrical about the Q-Q′ axis, and (b) the right two quadrants are symmetrical with the left two quadrants. (Naturally, in this latter case, the time-delays are reversed, with the left-ear signal leading the right-ear signal, rather than lagging it).

Consequently, it will be appreciated that there are two complementary positions in the horizontal plane associated with any particular (valid) time delay, for example 30° & 150°; 40° & 140°, and so on. In practice, measurements show that the time-delays are not truly symmetrical, and indicate, for example, that the maximum time delay occurs not at 90° azimuth, but at around 85°. These small asymmetries will be set aside for the moment, for clarity of description, but it will be seen that use of the time-delay as an index for the algorithm takes into account all of the detailed non-symmetries, thus providing a faithful means of simulating close sound sources.

Following on from this, if one considers the head as an approximately spherical object, one can see that the symmetry extends into the third dimension, where the upper hemisphere is symmetrical to the lower one, mirrored around the

horizontal plane. Accordingly, it can be appreciated that, for a given (valid) inter-aural time-delay, there exists not just a pair of points on the horizontal (h-) plane, but a locus, approximately circular, which intersects the h-plane at the aforementioned points. In fact, the locus can be depicted as the surface of an imaginary cone, extending from the appropriate listener's ear, aligned with the lateral axis Q-Q′ (Figures 3 and 4).

At this stage, it is important to note that:

(1) the inter-aural time-delay represents a very close approximation of the relative acoustic path length difference between a sound source and each of the ears; and

(2) the inter-aural time-delay is an integral feature of every HRTF pair.

Consequently, when any 3D-sound synthesis system is using HRTF data, the associated inter-aural time delay can be used as an excellent index of relative path length difference. Because it is based on physical measurements, it is therefore a true measure, incorporating the various real-life non-linearities described above.

The next stage is to find out a means of determining the value of the signal gains which must be applied to the left and right-ear channels when a "close" virtual sound source is required. This can be done if the near- and far-ear situations are considered in turn, and if we use the 1 metre distance as the outermost reference datum, at which point we define the sound intensity to be 0 dB.

Figure 5 shows a plan view of the listener's head, together with the near-field area surrounding it. In the first instance, we are particularly interested in the front-right quadrant. If we can define a relationship between near-field spatial position in the h-plane and distance to the near-ear (right ear in this case), then this can be used to control the right-channel gain. The situation is trivial to resolve, as shown in Figure 6, if the "true" source-to-ear paths for the close frontal positions (such as path "A") are assumed to be similar to the direct distance (indicated by "B"). This simplifies the situation, as is shown on the left diagram of

Figure 6, indicating a sound source S in the front-right quadrant, at an azimuth angle of θ with respect to the listener. Also shown is the distance, d, of the sound source from the head centre, and the distance, p, of the sound source from the near-ear. The angle subtended by S-head-Q′ is (90° - θ). The near-ear distance can be derived using the cosine rule, from triangle S-head_centre-near_ear:

$$p^2 = d^2 + r^2 - 2dr.\cos(90-\theta)\Big|_{\theta=0}^{\theta=90} \tag{2}$$

If we assume the head radius, r, is 7.5 cm, then p is given by:

$$p = \sqrt{d^2 + (7.5)^2 - 15d.\sin\theta}\,\Big|_{\theta=0}^{\theta=90} \tag{3}$$

Figure 7 shows a plan view of the listener's head, together with the near-field area surrounding it. Once again, we are particularly interested in the front-right quadrant. However, the path between the sound source and the far-ear comprises two serial elements, as is shown clearly in the right hand detail of Figure 7. First, there is a direct path from the source, S, tangential to the head, labelled q, and second, there is a circumferential path around the head, C, from the tangent point, T, to the far-ear. As before, the distance from the sound source to the centre of the head is d, and the head radius is r. The angle subtended by the tangent point and the head centre at the source is angle R.

The tangential path, q, can be calculated simply from the triangle:

$$q = \sqrt{\left(d^2 - r^2\right)} \tag{4}$$

and also the angle R:

$$R = \sin^{-1}\left(\frac{r}{d}\right) \tag{5}$$

Considering the triangle S-T-head_centre, the angle P-head_centre-T is (90 - θ - R), and so the angle T-head_centre-Q (the angle subtended by the arc itself) must be (θ + R). The circumferential path can be calculated from this angle, and is:

$$C = \left\{\frac{\theta + R}{360}\right\} 2\pi r \qquad (6)$$

Hence, by substituting (5) into (6), and combining with (4), an expression for the total distance (in cm) from sound source to far-ear for a 7.5 cm radius head can be calculated:

$$\text{Far-Ear Total Path} = \sqrt{\left(d^2 - 7.5^2\right)} + 2\pi r \left\{\frac{\theta + \sin^{-1}\left(\frac{7.5}{d}\right)}{360}\right\} \qquad (7)$$

It is instructive to compute the near-ear gain factor as a function of azimuth angle at several distances from the listener's head. This has been done, and is depicted graphically in Figure 10. The gain is expressed in dB units with respect to the 1 metre distance reference, defined to be 0 dB. The gain, in dB, is calculated according to the inverse square law from path length, d (in cm), as:

$$\text{gain (dB)} = 10 \log\left(\frac{10^4}{d^2}\right) \qquad (8)$$

As can be seen from the graph, the 100 cm line is equal to 0 dB at azimuth 0°, as one expects, and as the sound source moves around to the 90° position, in line with the near-ear, the level increases to +0.68 dB, because the source is actually slightly closer. The 20 cm distance line shows a gain of 13.4 dB at azimuth 0°, because, naturally, it is closer, and, again, the level increases as the sound source moves around to the 90° position, to 18.1: a much greater increase this time.

The other distance lines show intermediate properties between these two extremes.

Next, consider the near-ear gain factor. This is depicted graphically in Figure 11. As can be seen from the graph, the 100 cm line is equal to 0 dB at azimuth 0° (as one expects), but here, as the sound source moves around to the 90 position, away from the far-ear, the level decreases to -0.99 dB. The 20 cm distance line shows a gain of 13.8 dB at azimuth 0°, similar to the equidistant near-ear, and, again, the level decreases as the sound source moves around to the 90 position, to 9.58: a much greater decrease than for the 100 cm data. Again, the other distance lines show intermediate properties between these two extremes.

It has been shown that a set of HRTF gain factors suitable for creating near-field effects for virtual sound sources can be calculated, based on the specified azimuth angle and required distance. However, in practice, the positional data is usually specified in spherical co-ordinates, namely: an angle of azimuth, θ, and an angle of elevation, ϕ (and now, according to the invention, distance, d). Accordingly, it is required to compute and transform this data into an equivalent h-plane azimuth angle (and in the range 0° to 90°) in order to compute the appropriate L and R gain factors, using equations (3) and (7). This can require significant computational resource, and, bearing in mind that the CPU or dedicated DSP will be running at near-full capacity, is best avoided if possible.

An alternative approach would be to create a universal "look-up" table, featuring L and R gain factors for all possible angles of azimuth and elevation (typically around 1,111 in an HRTF library), at several specified distances. Hence this table, for four specified distances, would require 1,111 x 4 x 2 elements (8,888), and therefore would require a significant amount of computer memory allocated to it.

The inventors have, however, realised that the time-delay carried in each HRTF can be used as an index for selecting the appropriate L and R gain factors. Every inter-aural time-delay is associated with a horizontal plane equivalent, which, in turn, is associated with a specific azimuth angle. This means that a much smaller look-up table can be used. An HRTF library of the above resolution

features horizontal plane increments of 3°, such that there are 31 HRTFs in the range 0° to 90°. Consequently, the size of a time-delay-indexed look-up table would be 31 x 4 x 2 elements (248 elements), which is only 2.8% the size of the "universal" table, above.

The final stage in the description of the invention is to tabulate measured, horizontal-plane, HRTF time-delays in the range 0° to 90° against their azimuth angles, together with the near-ear and far-ear gain factors derived in previous sections. This links the time-delays to the gain factors, and represents the look-up table for use in a practical system. This data is shown below in the form of Table 1 (near-ear data) and Table 2 (far-ear data).

Time-Delay (samples)	Azimuth (degrees)	d = 20 (cm)	d = 40 (cm)	d = 60 (cm)	d = 80 (cm)	d = 100 (cm)
0	0	13.41	7.81	4.37	1.90	-0.02
1	3	13.56	7.89	4.43	1.94	0.01
2	6	13.72	7.98	4.48	1.99	0.04
4	9	13.88	8.06	4.54	2.03	0.08
5	12	14.05	8.15	4.60	2.07	0.11
6	15	14.22	8.24	4.66	2.11	0.15
7	18	14.39	8.32	4.71	2.16	0.18
8	21	14.57	8.41	4.77	2.20	0.21
9	24	14.76	8.50	4.83	2.24	0.25
10	27	14.95	8.59	4.88	2.28	0.28
11	30	15.14	8.68	4.94	2.32	0.31
12	33	15.33	8.76	4.99	2.36	0.34
13	36	15.53	8.85	5.05	2.40	0.37
14	39	15.73	8.93	5.10	2.44	0.40
15	42	15.93	9.01	5.15	2.48	0.43
16	45	16.12	9.09	5.20	2.51	0.46
18	48	16.32	9.17	5.25	2.55	0.49
19	51	16.51	9.24	5.29	2.58	0.51
20	54	16.71	9.32	5.33	2.61	0.53
21	57	16.89	9.38	5.37	2.64	0.56
23	60	17.07	9.44	5.41	2.66	0.58
24	63	17.24	9.50	5.44	2.69	0.59
25	66	17.39	9.55	5.48	2.71	0.61
26	69	17.54	9.60	5.50	2.73	0.63
27	72	17.67	9.64	5.53	2.74	0.64
27	75	17.79	9.68	5.55	2.76	0.65
28	78	17.88	9.71	5.57	2.77	0.66
28	81	17.96	9.73	5.58	2.78	0.67
29	84	18.02	9.75	5.59	2.79	0.67
29	87	18.05	9.76	5.59	2.79	0.68
29	90	18.06	9.76	5.60	2.79	0.68

Table 1
Time-delay based look-up table for determining near-ear gain factor as function of distance between virtual sound source and centre of the head.

Time-Delay (samples)	Azimuth (degrees)	d = 20 (cm)	d = 40 (cm)	d = 60 (cm)	d = 80 (cm)	d = 100 (cm)
0	0	13.38	7.81	4.37	1.90	-0.02
1	3	13.22	7.72	4.31	1.86	-0.06
2	6	13.07	7.64	4.26	1.82	-0.09
4	9	12.92	7.56	4.20	1.77	-0.13
5	12	12.77	7.48	4.15	1.73	-0.16
6	15	12.62	7.40	4.09	1.69	-0.19
7	18	12.48	7.32	4.04	1.65	-0.23
8	21	12.33	7.24	3.98	1.61	-0.26
9	24	12.19	7.16	3.93	1.57	-0.29
10	27	12.06	7.08	3.88	1.53	-0.33
11	30	11.92	7.01	3.82	1.49	-0.36
12	33	11.79	6.93	3.77	1.45	-0.39
13	36	11.66	6.86	3.72	1.41	-0.42
14	39	11.53	6.78	3.67	1.37	-0.46
15	42	11.40	6.71	3.61	1.33	-0.49
16	45	11.27	6.63	3.56	1.29	-0.52
18	48	11.15	6.56	3.51	1.25	-0.55
19	51	11.03	6.49	3.46	1.21	-0.58
20	54	10.91	6.42	3.41	1.17	-0.62
21	57	10.79	6.35	3.36	1.13	-0.65
23	60	10.67	6.27	3.31	1.09	-0.68
24	63	10.55	6.20	3.26	1.05	-0.71
25	66	10.44	6.14	3.21	1.01	-0.74
26	69	10.33	6.07	3.16	0.97	-0.77
27	72	10.22	6.00	3.11	0.94	-0.80
27	75	10.11	5.93	3.06	0.90	-0.84
28	78	10.00	5.86	3.01	0.86	-0.87
28	81	9.89	5.80	2.97	0.82	-0.90
29	84	9.78	5.73	2.92	0.79	-0.93
29	87	9.68	5.66	2.87	0.75	-0.96
29	90	9.58	5.60	2.82	0.71	-0.99

Table 2
Time-delay based look-up table for determining far-ear gain factor as function of distance between virtual sound source and centre of the head.

Note that the time-delays in the above tables are shown in units of sample periods related to a 44.1 kHz sampling rate, hence each sample unit is 22.676 μs.

Consider, by way of example, the case when a virtual sound source is required to be positioned in the horizontal plane at an azimuth of 60°, and at a distance of 0.4 metres. Using Table 1, the near-ear gain which must be applied to the HRTF is shown as 9.44 dB. and the far-ear gain (from Table 2) is 6.27 dB.

Consider, as a second example, the case when a virtual sound source is required to be positioned out of the horizontal plane, at an azimuth of 42° and elevation of -60°, at a distance of 0.2 metres. The HRTF for this particular spatial position has a time-delay of 7 sample periods (at 44.1 kHz). Consequently, using Table 1, the near-ear gain which must be applied to the HRTF is shown as 14.39 dB, and the far-ear gain (from Table 2) is 12.48 dB. (This HRTF time-delay is the same as that of a horizontal-plane HRTF with an azimuth value of 18°).

The implementation of the invention is straightforward, and is depicted schematically in Figure 9. Figure 8 shows the conventional means of creating a virtual sound source, as follows. First, the spatial position of the virtual sound source is specified, and used to select an HRTF appropriate to that position. The HRTF comprises a left-ear function, a right-ear function and an inter-aural time-delay value. In a computer system for creating the virtual sound source, the HRTF data will generally be in the form of FIR filter coefficients suitable for controlling a pair of FIR filters (one for each channel), and the time-delay will be represented by a number. A monophonic sound source is then transmitted into the signal-processing scheme, as shown, thus creating both a left- and right-hand channel outputs. (These output signals are then suitable for onward transmission to the listener's headphones, or crosstalk-cancellation processing for loudspeaker reproduction, or other means).

The invention, shown in Figure 9, supplements this procedure, but requires little extra computation. This time, the signals are processed as previously, but a near-field distance is also specified, and, together with the time-delay data from the selected HRTF, is used to select the gain for respective left and right channels

from a look-up table; this data is then used to control the gain of the signals before they are output to subsequent stages, as described before.

The left channel output and the right channel output shown in Figure 9 can be combined directly with a normal stereo or binaural signal being fed to headphones, for example, simply by adding the signal in corresponding channels. If the outputs shown in Figure 9 are to be combined with those created for producing a 3D sound-field generated, for example, by binaural synthesis (such as, for example, using the Sensaura (Trade Mark) method described in EP-B-0689756), then the two output signals should be added to the corresponding channels of the binaural signal after transaural crosstalk compensation has been performed.

Although in the example described above the setting of magnitude of the left and right signals is performed after modification using a head response transfer function, the magnitudes may be set before such signal processing if desired, so that the order of the steps in the described method is not an essential part of the invention.

Although in the example described above the position of the virtual sound source relative to the preferred position of a listener's head in use is constant and does not change with time, by suitable choice of sucessive different positions for the virtual sound source it can be made to move relative to the head of the listener in use if desired. This apparent movement may be provided by changing the direction of the virtual souce from the preferred position, by changing the distance to it, or by changing both together.

Finally, the content of the accompanying abstract is hereby incorporated into this description by reference.

CLAIMS

1. A method of processing a single channel audio signal to provide an audio signal having left and right channels corresponding to a sound source at a given direction in space relative to a preferred position of a listener in use, the information in the channels including cues for perception of the direction of said single channel audio signal from said preferred position, the method including the steps of: a) providing a two channel signal having the same single channel signal in the two channels; b) modifying the two channel signal by modifying each of the channels using one of a plurality of head response transfer functions to provide a right signal in one channel for the right ear of a listener and a left signal in the other channel for the left ear of the listener; and c) introducing a time delay between the channels corresponding to the inter-aural time difference for a signal coming from said given direction, the inter-aural time difference providing cues to perception of the direction of the sound source at a given time, characterised in that the method includes controlling the magnitude of the left signal and the right signal to be at respective values at said given time, the values being chosen to provide cues for perception of the distance of said sound source from said preferred position at said given time.

2. A method of processing a single channel audio signal as claimed in claim 1 in which the left signal magnitude and the right signal magnitude are chosen separately.

3. A method as claimed in any preceding claim in which the left ear signal magnitude and right ear signal magnitude are determined by choosing a position for the sound source relative to said preferred position of the head of a listener in use, determining the distance from the chosen position of the sound source to respective ears of said listener, and determining the corresponding left signal magnitude and right signal magnitude using the inverse square law dependence of sound intensity with distance.

4. A method as claimed in claim 3 in which the distance from the chosen position of the sound source, at said given time, to respective ears of said listener is determined from a look-up table.

5. A method as claimed in claim 3 in which the distance from the position of the sound source, at said given time, to the centre of the head of said listener is chosen, and the distance to respective ears is determined from the inter-aural time delay.

6. A method as claimed in claim 5 in which the distance to respective ears is determined from a look-up table.

7. A method as claimed in any preceding claim in which the magnitude of the left signal or the magnitude of the right signal is sufficiently small as to be inaudible.

8. A method as claimed in any preceding claim in which the left signal and right signal are compensated to cancel or reduce transaural crosstalk when supplied as left and right channels for replay by loudspeakers.

9. A method as claimed in any preceding claim in which the resulting two channel audio signal is combined with a further two or more channel audio signal.

10. A method as claimed in claim 9 in which the signals are combined by adding the content of corresponding channels to provide a combined signal having two channels.

11. A computer program for implementing a method as claimed in any preceding claim.

12. Apparatus for performing the method as claimed in any preceding claim.

13. An audio signal processed by a method as claimed in any of claims 1 - 10.

Fig.1.

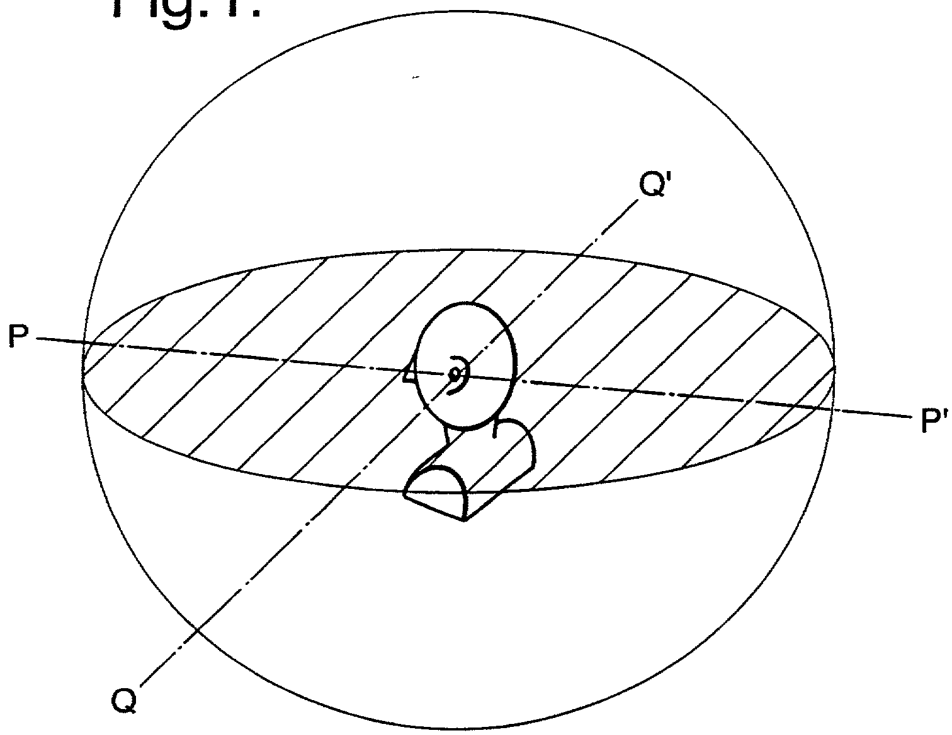


Fig.2.

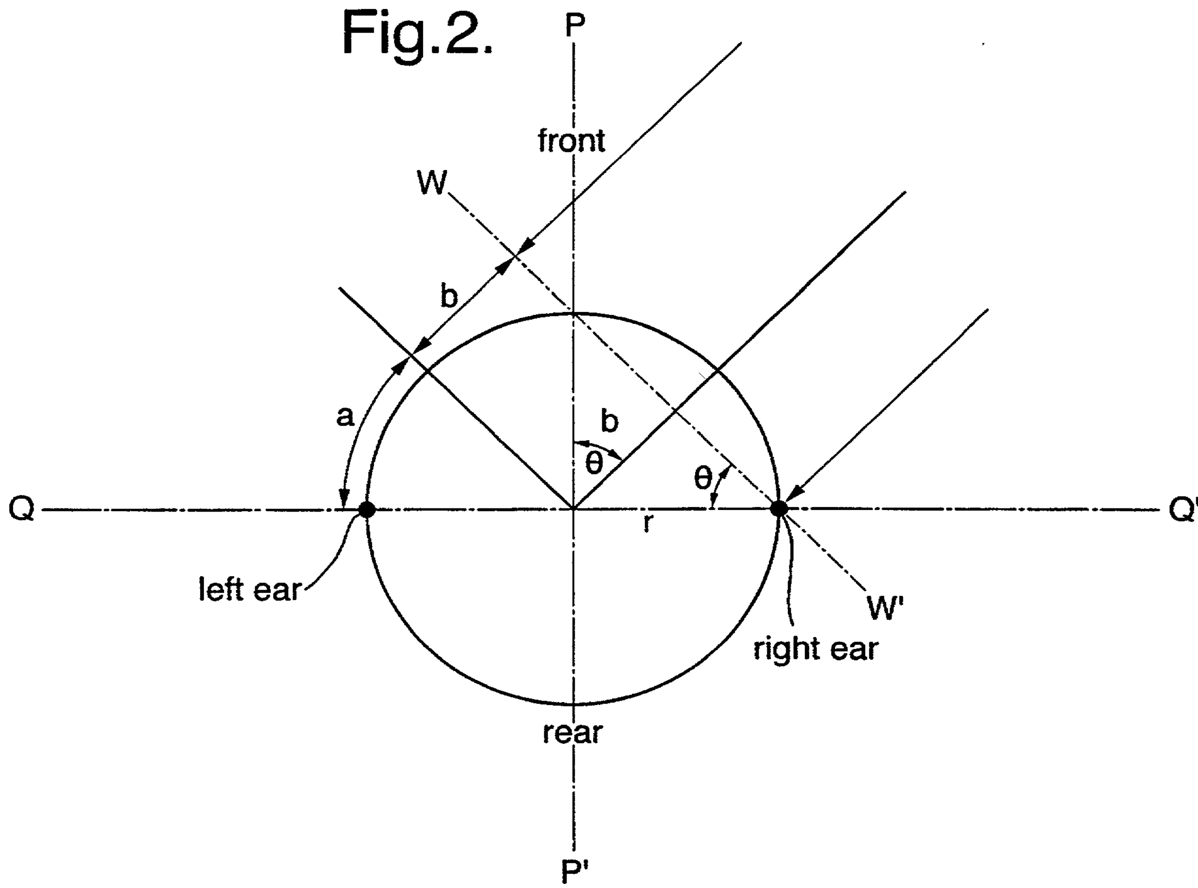


Fig.3.

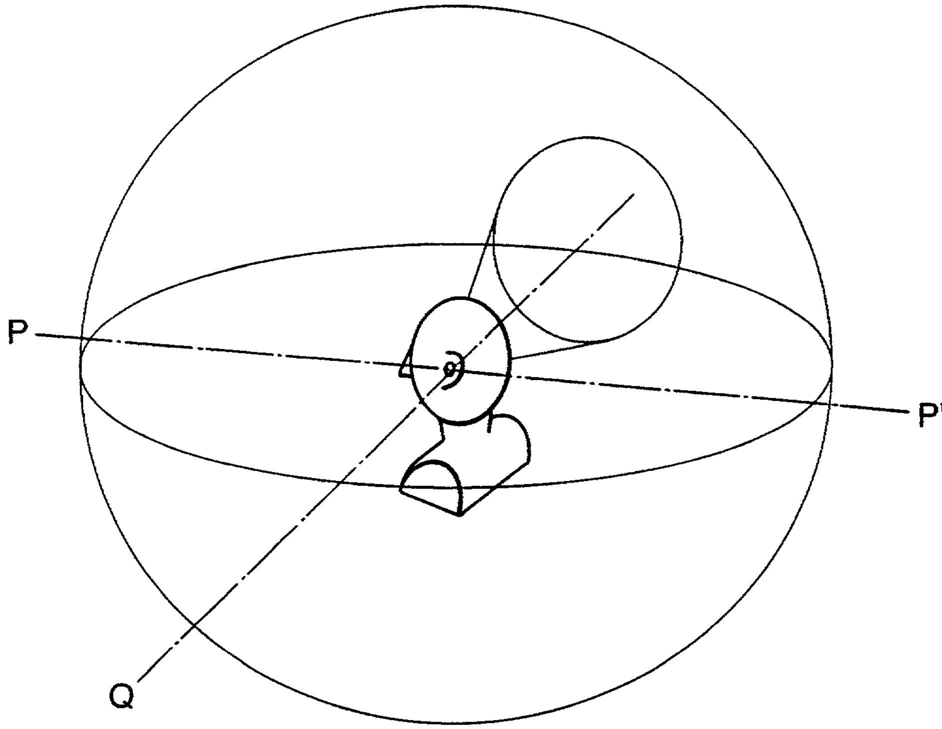


Fig.4.

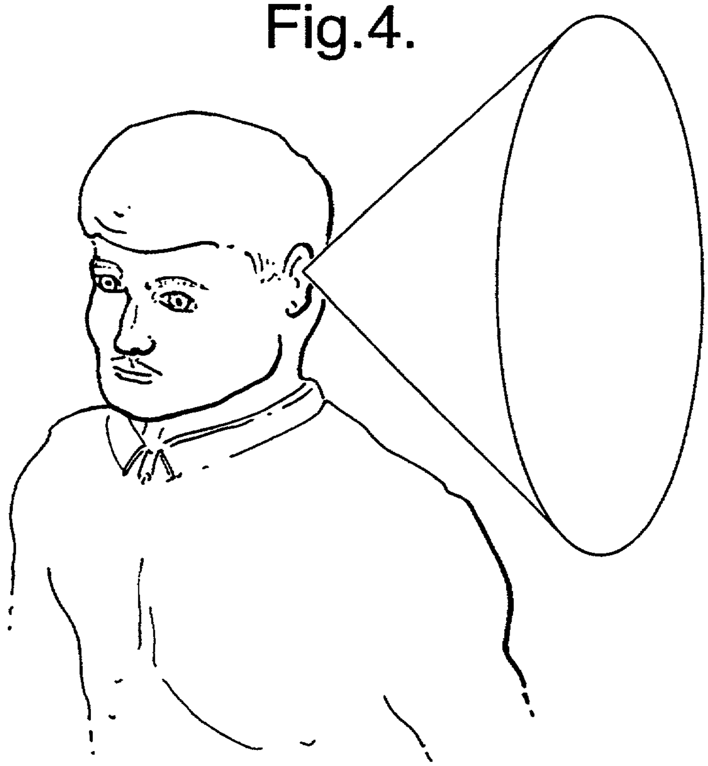
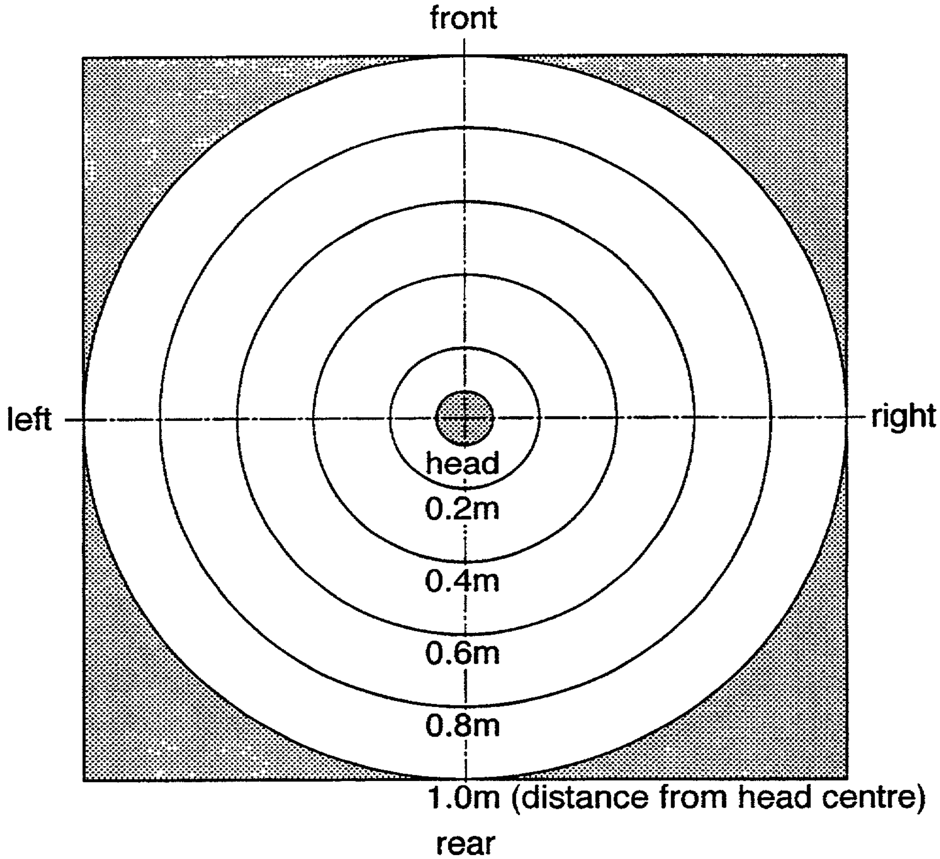


Fig.5.


front
left
right
head
0.2m
0.4m
0.6m
0.8m
1.0m (distance from head centre)
rear

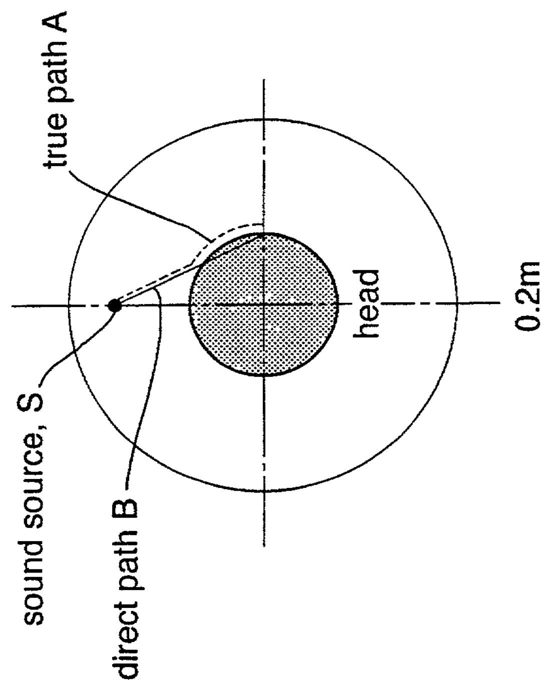

true path A
sound source, S
direct path B
head
0.2m

Fig.6.

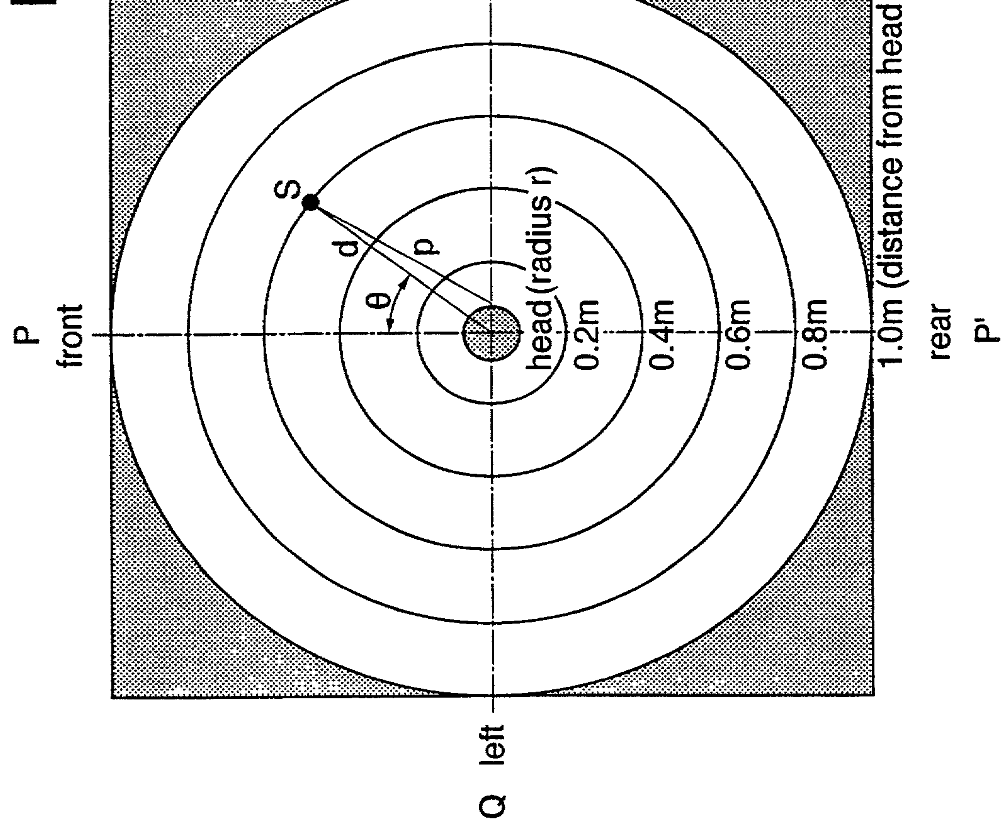
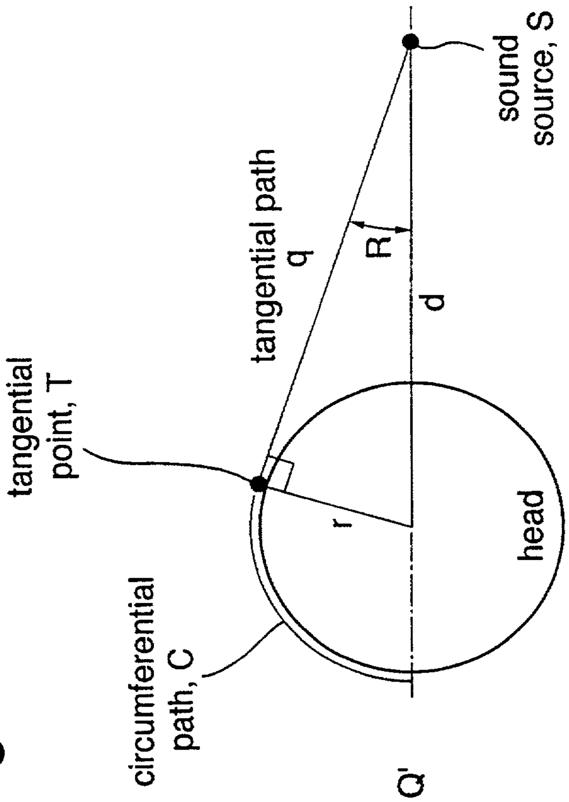
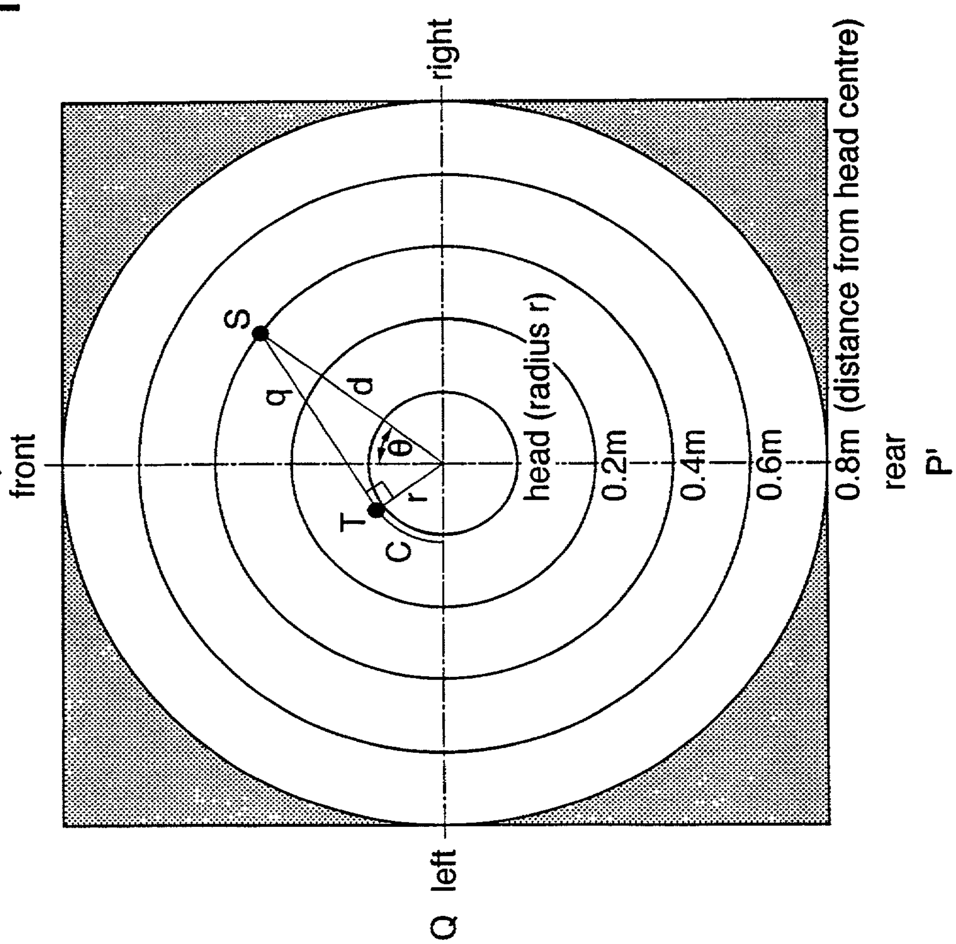

P
front
Q'
right
Q
left
P'
rear
S
d
p
θ
head (radius r)
0.2m
0.4m
0.6m
0.8m
1.0m (distance from head centre)

Fig.7.



tangential point, T
tangential path
q
R
d
sound source, S
r
head
circumferential path, C



P
front
Q left
right Q'
S
q
d
θ
T
r
C
head (radius r)
0.2m
0.4m
0.6m
0.8m (distance from head centre)
rear
P'

Fig.8.

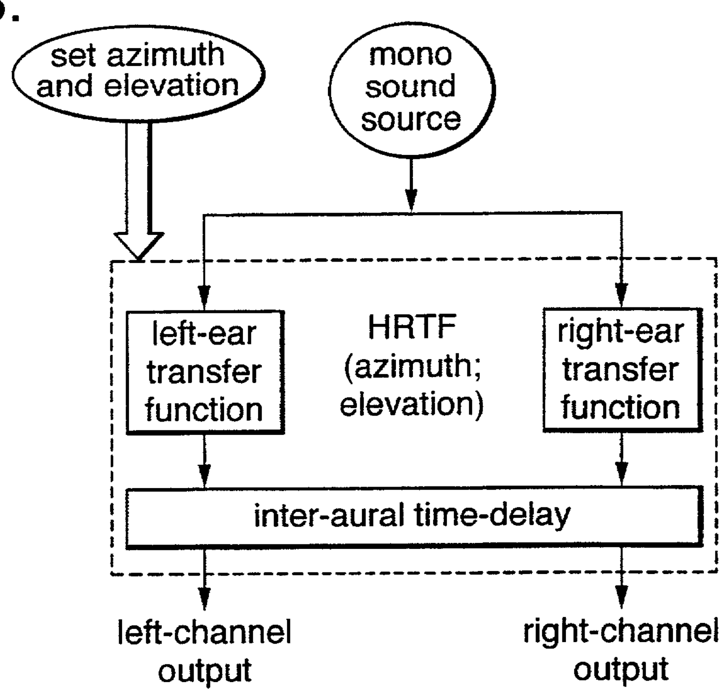

set azimuth and elevation
mono sound source
left-ear transfer function
HRTF (azimuth; elevation)
right-ear transfer function
inter-aural time-delay
left-channel output
right-channel output

Fig.9.

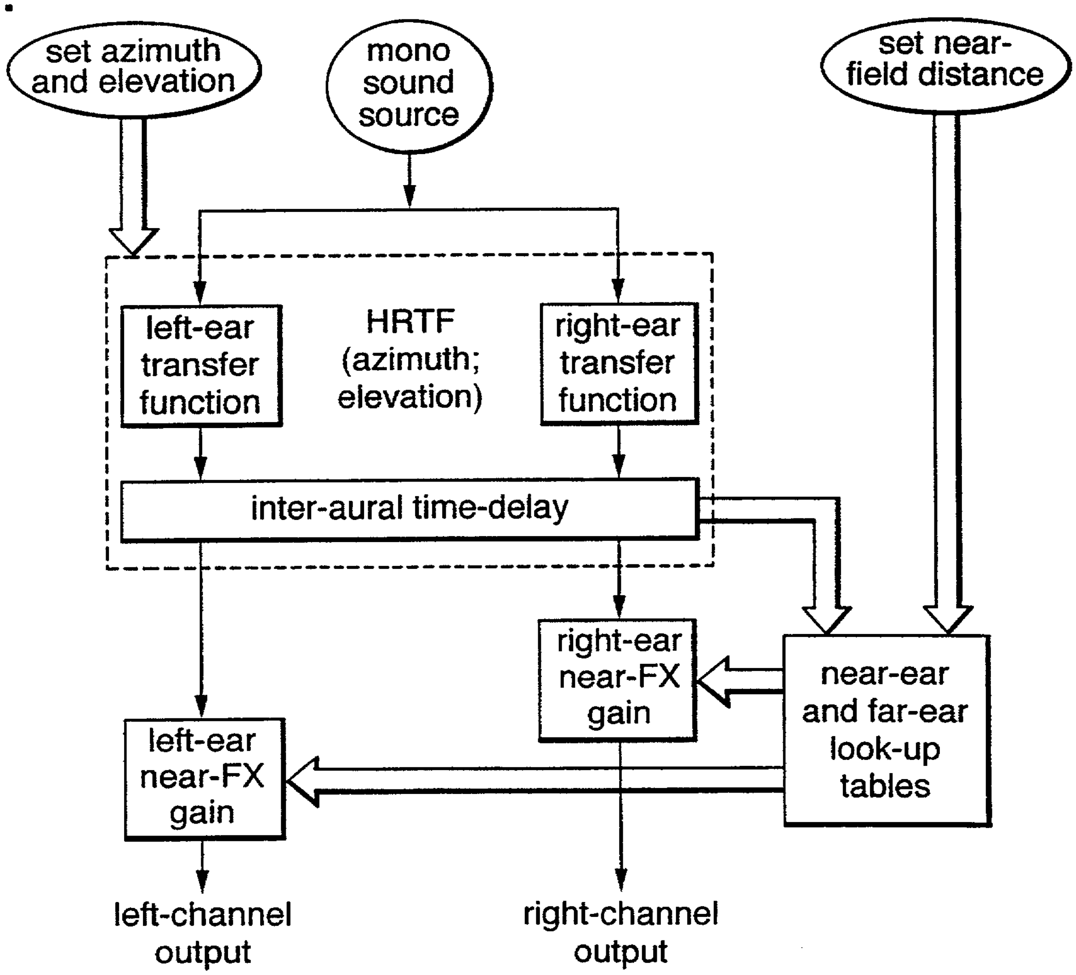

set azimuth and elevation
mono sound source
set near-field distance
left-ear transfer function
HRTF (azimuth; elevation)
right-ear transfer function
inter-aural time-delay
right-ear near-FX gain
near-ear and far-ear look-up tables
left-ear near-FX gain
left-channel output
right-channel output

Fig.10.

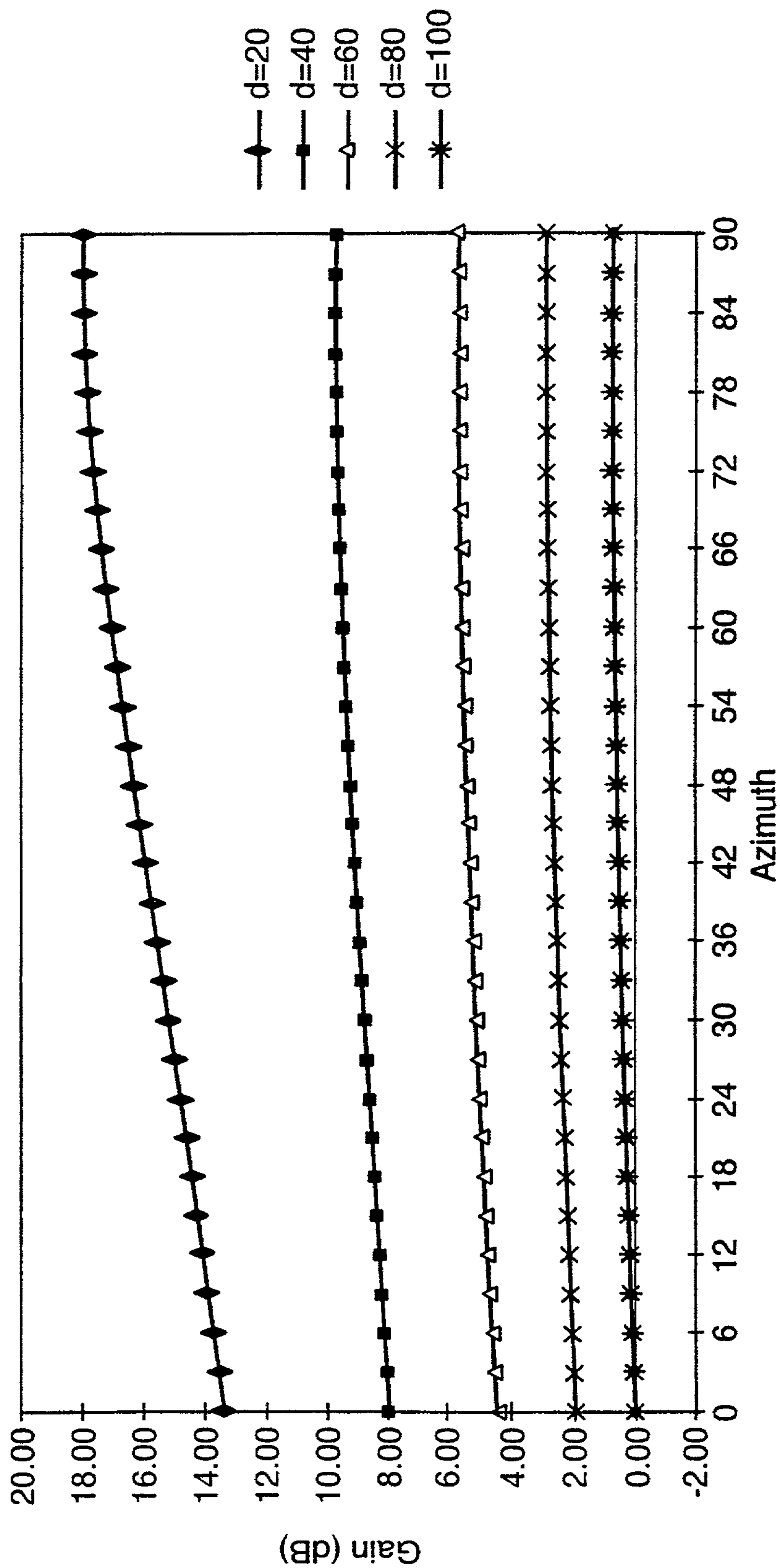
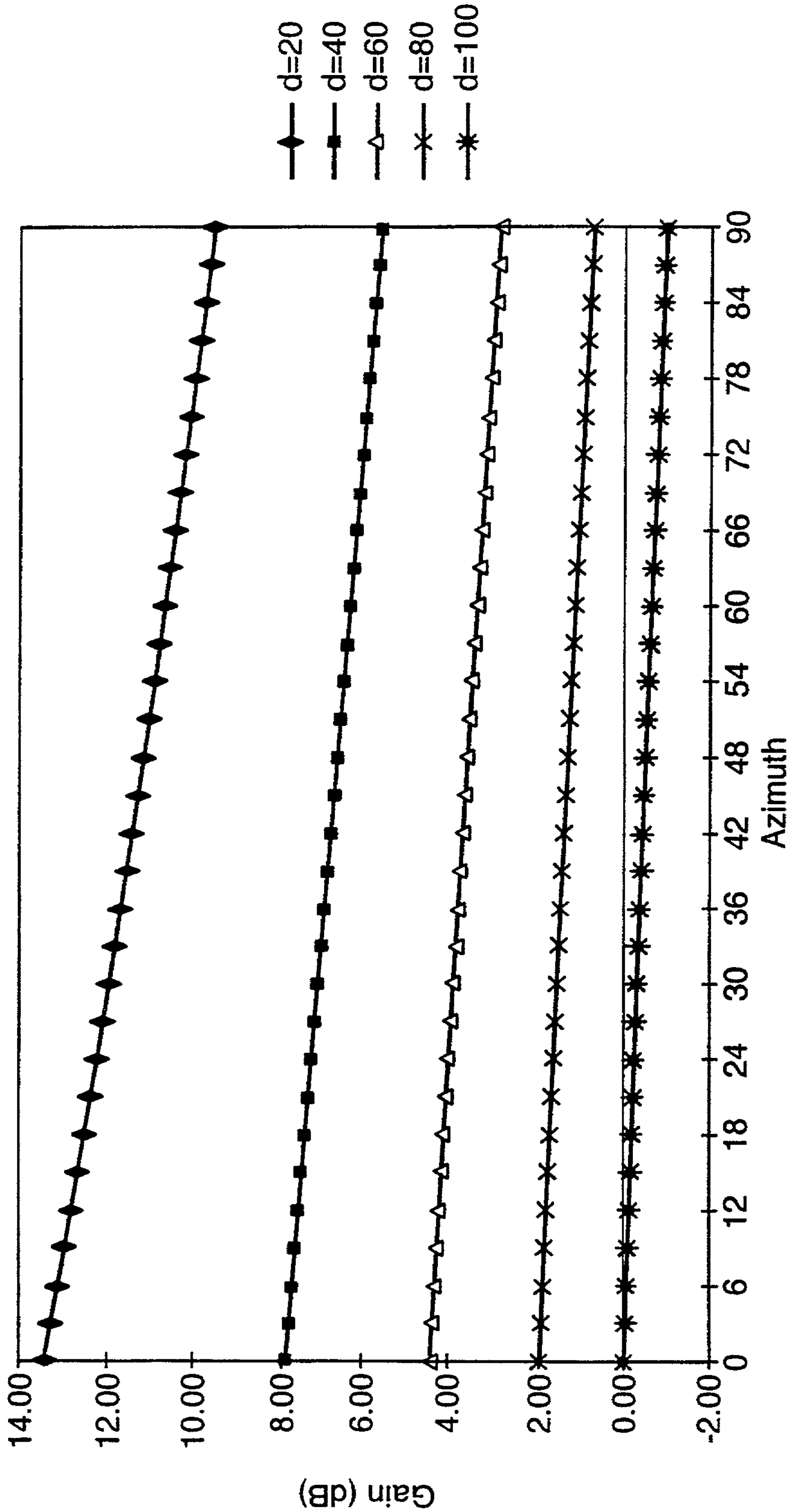

d=20
d=40
d=60
d=80
d=100
Gain (dB)
20.00
18.00
16.00
14.00
12.00
10.00
8.00
6.00
4.00
2.00
0.00
-2.00
Azimuth
0
6
12
18
24
30
36
42
48
54
60
66
72
78
84
90

Fig.11.


d=20
d=40
d=60
d=80
d=100
Gain (dB)
14.00
12.00
10.00
8.00
6.00
4.00
2.00
0.00
-2.00
0
6
12
18
24
30
36
42
48
54
60
66
72
78
84
90
Azimuth

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

Declaration and Power of Attorney

As the below named inventors, we hereby declare that:

Our residence, post office address and citizenship are as stated below next to our names.

We believe we are the original, first and joint inventors of the subject matter of which is claimed and for which a patent is sought on the invention entitled **A METHOD OF PROCESSING AN AUDIO SIGNAL** the specification of which was filed as PCT international Application Number PCT/GB98/03714 filed December 11, 1998, and which is being amended by a preliminary amendment being filed concurrently herewith.

We hereby state that we have reviewed and understand the contents of the above identified application, including the specification and the claims, as amended by an amendment, if any, specifically referred to in this oath or declaration.

We acknowledge the duty to disclose all information known to us which is material to patentability as defined in Title 37, Code of Federal Regulations, 1.56.

We hereby claim foreign priority benefits under Title 35, United States Code, 119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

British Patent Application No. 9726338.8 filed December 13, 1997, and entitled "A METHOD OF PROCESSING AN AUDIO SIGNAL".

We hereby claim the benefit under Title 35, United States Code, 120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, 112, We acknowledge the duty to disclose all information known to us to be material to patentability as defined in Title 37, Code of Federal Regulations, 1.56 which became available between the filing date of the prior application and the national or PCT international filing date of this application:

None

We hereby declare that all statements made herein of our own knowledge are true and that all statements made on information and belief are believed to be true; and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

We hereby appoint the following attorneys with full power of substitution and revocation, to prosecute said application, to make alterations and amendments therein, to receive the patent, and to transact all business in the Patent and Trademark Office connected therewith:

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